

IN THE SPECIFICATION

At page 4, first paragraph, lines 9 to 10 and line 13:

Most speech coding systems in use today are based on telephone-bandwidth narrowband speech, nominally limited to about 200-3400 Hz and sampled at a rate of 8 kHz. The inherent bandwidth limitations cause degradation to the communication quality. Recently, there are various efforts to develop wideband speech (band-limited to about 20 ~ 7000 Hz) coding systems surpassing the quality of conventional telephone-bandwidth speech. The 3rd Generation Partnership Project (3GPP) and the International Telecommunication Union-Telecommunication (ITU-T) have recognized the importance of wideband speech and had selected the Adaptive Multi Rate - WideBand (AMR-WB), a.k.a. and ITU-T G.722.2 as their wideband speech codec standard. And also the 3rd Generation Partnership Project 2 (3GPP2) goes through with its own wideband speech codec standard. Thus narrowband speech network networks and wideband speech network networks may co-exist in the near future. When networks employing the different codec standard are inter-networking through the gateway system, there is a need for translation of the coded bit stream. Generally, when we interlink the networks employing the different codecs with the different bandwidths, we need more sophisticated translation skill. This translation operation is so called (trans-coding.)” The conventional and simple solution is that an encoder part of one codec is concatenated to a decoder part of the other codec.

At page 8, fourth paragraph, lines 17 to 18:

The formant parameter translator 32 translates α-formant parameters encoded in an input CELP format into an output CELP format and generates formant parameters in the output CELP format.